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BEFORE THE BOARD OF PATENT APPEALS
AND INTERFERENCES

Application Number: 10/077,405
Filing Date: February 15, 2002
Appellant(s): LEBLANC, WILFRID

John A. WIlberg
For Appellant

EXAMINER'S ANSWER

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John A. Wiberg
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EXAMINER'S ANSWER

This is in response to the appeal brief filed August 28, 2008 appealing from the Office action mailed November 27, 2008.

(1) Real Party in Interest

A statement identifying by name the real party in interest is contained in the brief.

(2) Related Appeals and Interferences

The examiner is not aware of any related appeals, interferences, or judicial proceedings which will directly affect or be directly affected by or have a bearing on the Board's decision in the pending appeal.

(3) Status of Claims

The statement of the status of claims contained in the brief is correct.

(4) Status of Amendments After Final

No amendment after final has been filed.

(5) Summary of Claimed Subject Matter

The summary of claimed subject matter contained in the brief is correct.

(6) Grounds of Rejection to be Reviewed on Appeal

The appellant's statement of the grounds of rejection to be reviewed on appeal is correct.

(7) Claims Appendix

The copy of the appealed claims contained in the Appendix to the brief is correct.

(8) Evidence Relied Upon

US 5,623,483	Agrawal et al.	4-1997
US 6,097,697	Yao et al.	8-2000
US 5,905,711	Chiussi et al.	5-1999
US 6,810,377	Ho et al.	10-2004

(9) Grounds of Rejection

The following ground(s) of rejection are applicable to the appealed claims:

Claim Rejections - 35 USC § 102

The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

1. Claim 1 is rejected under 35 U.S.C. 102(b) as being anticipated by Agrawal (US 5,623,483).

Regarding claim 1, Agrawal describes a method of processing digital media data stream comprising a stream of data elements (title, multimedia streams), comprising:

(a) receiving the data stream (fig. 1 & col. 2, lines 50-52, stream receiver 30);

(b) holding each data element that is received prior to an end of a time period in a buffer until the end of the time period, at which time the data element is released for playout (col. 5, lines 24-33, buffer control 200 holds each data stream element in buffer slot designated by a respective pointer until timer 200, after T_r seconds, signals the element to be moved to output device 40);

(c) monitoring a loss rate at which data elements in the data stream are not received by the end of their respective time periods (col. 6, lines 15-17, monitoring any changes to the PLR (packet loss rate) of the data stream in the buffer);

(d) adjusting a duration of the time period based upon the loss rate (col. 6, lines 15-17, updating (adjusting) the buffer operating characteristics described in col. 5, lines 24-39, including adjusting the timer 230 to a new T_r seconds delay packet rate).

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. Claims 2-6 and 9-11 are rejected under 35 U.S.C. 103(a) as being unpatentable over Agrawal (US 5,623,483) in view of Yao (6,097,697).

Regarding claim 2, Agrawal fails to describe:

adjusting step (d) comprises increasing the duration of the time period if the loss rate is above a first threshold.

Yao describes:

adjusting step (d) comprises increasing the duration of the time period if the loss rate is above a first threshold (col. 5, lines 8-15, “On loss rate axis 310, a loss hysteresis threshold [LOSS_HYST] 312 defines a range 314 between LOSS_HYST and 1.0. In this range, an excess loss rate contributes to a decrease in transmission rate” [i.e. increased duration of time period].)

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to incorporate the particular adjustment steps as a function of the loss rate as in Yao for the teaching of Agrawal.

The motivation for combining the teaching is that “The [loss rate] statistics provide indications of congestion of the data network” (Yao, col. 2, lines 56-57), and the statistic may be used to minimize such network congestion.

Regarding claim 3, Agrawal fails to explicitly describe:

adjusting step (d) comprises setting the duration of the time period at a first value if the loss rate is relatively low, and setting the duration at a second value, greater than the first value, if the loss rate is relatively higher

Yao describes:

adjusting step (d) comprises setting the duration of the time period at a first value (col. 6, line 26, new transmission rate R_new) if the loss rate is relatively low (col. 6, lines 23, “If the combined factor is negative, then the rate [R-new] is decrease”, where

the combined factor comprises two “loss rate” affecting (sub)-factors: “Based on the loss ratio and excess loss rate of a sequence of packets, rate controller 116 computes two factors, a span factor and a loss factor”, col. 5, lines 41-43);

and setting the duration at a second value (col. 6, line 28, new transmission rate R_new), greater than the first value (increased transmission rate), if the loss rate is relatively higher (col. 6, lines22- 23, “If the combined factor is positive, then the rate [R-new] is increased”, where the combined factor comprises two “loss rate” affecting (sub)-factors: “Based on the loss ratio and excess loss rate of a sequence of packets, rate controller 116 computes two factors, a span factor and a loss factor”, col. 5, lines 41-43).

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to incorporate the particular adjustment steps as a function of the loss rate as in Yao for the teaching of Agrawal.

The motivation for combining the teaching is that “The [loss rate] statistics provide indications of congestion of the data network” (Yao, col. 2, lines 56-57), and the statistic may be used to minimize such network congestion.

Regarding claim 4, Agrawal fails to explicitly describe:

adjusting step (d) comprises decreasing the duration of the time period if the loss rate is relatively low, and increasing the duration if the loss rate is relatively higher.

Yao describes:

adjusting step (d) comprises decreasing the duration of the time period if the loss rate is relatively low, and increasing the duration if the loss rate is relatively higher (col.

5, lines 8-15, "On loss rate axis 310, a loss hysteresis threshold [LOSS_HYST] 312 defines a range 314 between LOSS_HYST and 1.0. In this range, an excess (high) loss rate contributes to a decrease in transmission rate [i.e. increased duration of time period]. The negative of the loss hysteresis threshold [-LOSS_HYST] (low loss rate) 316 defines a range 318 from –LOSS_HYST to –1.0 in which the excess loss rate contributes to an increase in transmission rate [i.e. decreased duration of time period].")

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to incorporate the particular adjustment steps as a function of the loss rate as in Yao for the teaching of Agrawal.

The motivation for combining the teaching is that "The [loss rate] statistics provide indications of congestion of the data network" (Yao, col. 2, lines 56-57), and the statistic may be used to minimize such network congestion.

Regarding claim 5, Agrawal fails to describe:

(d)(i) if the loss rate is lower than a first threshold (LOSS_HYST), maintaining the duration of the time period at a present value, and

(d)(ii) if the loss rate is greater than the first threshold, increasing the duration of the time period by a first amount.

Yao describes the adjustment of step (d) comprises:

(d)(i) if the loss rate is lower than a first threshold (LOSS_HYST), maintaining the duration of the time period at a present value (fig. 3, range between #316 and #312, where transmission rate is unchanged) and

(d)(ii) if the loss rate is greater than the first threshold, increasing the duration of the time period by a first amount higher (col. 5, lines 8-15, "On loss rate axis 310, a loss hysteresis threshold [LOSS_HYST] 312 defines a range 314 between LOSS_HYST and 1.0. In this range, an excess (high) loss rate contributes to a decrease in transmission rate [i.e. increased duration of time period]").

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to incorporate the particular adjustment steps as a function of the loss rate as in Yao for the teaching of Agrawal.

The motivation for combining the teaching is that "The [loss rate] statistics provide indications of congestion of the data network" (Yao, col. 2, lines 56-57), and the statistic may be used to minimize such network congestion.

Regarding claim 6, Agrawal and Yao combined describe all limitations in claim 5. Agrawal and Yao further describe that step (d)(ii) comprises:

increasing the duration of the time period by a first amount that is substantially equivalent to a duration of the media represented by one data element (Agrawal, col. 6, lines 25-29, repeating a data element 1x to increase time period equivalent to a data element).

Regarding claim 9, Agrawal fails to describe:

(d)(i) if the loss rate is lower than a first threshold, decreasing the duration of the time period;

(d)(ii) if the loss rate is greater than the first threshold but less than a second threshold, and

(d)(iii) if the loss rate is greater than the second threshold (LOSS_HYST), increasing the duration of the time period.

Yao describes that step (d) comprises:

(d)(i) if the loss rate is lower than a first threshold (fig. 3, -LOSS_HYST #316), decreasing the duration of the time period;

(d)(ii) if the loss rate is greater than the first threshold but less than a second threshold (fig. 3, LOSS_HYST #312), maintaining the duration of the time period at a present value [fig. 3, between -LOSS_HYST #316 and LOSS_HYST #312); and

(d)(iii) if the loss rate is greater than the second threshold (LOSS_HYST), increasing the duration of the time period;

(col. 5, lines 8-15, "On loss rate axis 310, a loss hysteresis threshold [LOSS_HYST] 312 defines a range 314 between LOSS_HYST and 1.0. In this range, an excess (high) loss rate contributes to a decrease in transmission rate [i.e. increased duration of time period]. The negative of the loss hysteresis threshold [-LOSS_HYST] (low loss rate) 316 defines a range 318 from –LOSS_HYST to –1.0 in which the excess loss rate contributes to an increase in transmission rate [i.e. decreased duration of time period].")

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to incorporate the particular adjustment steps as a function of the loss rate as in Yao for the teaching of Agrawal.

The motivation for combining the teaching is that “The [loss rate] statistics provide indications of congestion of the data network” (Yao, col. 2, lines 56-57), and the statistic may be used to minimize such network congestion.

Regarding claim 10, Agrawal further describe that the data elements are frames of encoded data (col. 1, line 13, audio/video encoded frames).

Regarding claim 11, Agrawal suggests that the time period begins for each transmitted data element when the data element is sent by a transmitting end (col. 5, lines 46-47, in using the timestamp of the sender).

3. Claims 7 and 8 are rejected under 35 U.S.C. 103(a) as being unpatentable over Agrawal in view of Yao as applied to claim 5 above, and further in view of Chiussi (5,905,711).

Regarding claim 7, Agrawal and Yao combined describe all limitations in claim 5. Agrawal further suggests the adjustment of step (d) comprises:

(d)(iii) if the loss rate is greater than a threshold, increasing the duration of the time period by a second amount that is greater than the first amount (col. 25-29, where the playout duration is increased by a second amount equivalent to repeatedly transmitting a data packet/element X times, which is X times greater in amount than repeatedly transmitting a data packet/element 1 time).

Agrawal fails to describe:

a second threshold that is greater than the first threshold (abstract, where second threshold is a greater value than the first threshold.

Chiussi describe:

a second threshold that is greater than the first threshold (abstract, where second threshold is a greater value than the first threshold [to direct **all** data sources to reduce data transfer rate).

It would have been obvious to one of ordinary skill in the art at the time or motivation to describe first and second thresholds to indicate the step increase of duration of time period in Agrawal and Yao.

The motivation for combining the teachings is that this is “a method and apparatus that achieves good performance by guaranteeing fairness and control on the buffer size and is simple to implement”, (col. 2, lines 12-14).

Regarding claim 8, Agrawal, Yao and Chiussi describe all limitations set forth in claim 7. Agrawal further suggests step (d)(ii) comprises:

increasing the duration of the time period by a first amount that is substantially equivalent to a duration of the media represented by one data element and wherein the second amount is substantially equivalent to twice the duration of the media represented by one data element ((Agrawal, col. 6, lines 25-29, repeating a data element 1x to increase time period equivalent to a data packet/element, or repeating a data element 2x to increase the duration equivalent to 2 bytes of data packet/element).

4. Claims 12-14, 21, 23-25, 30 and 32-33 are rejected under 35 U.S.C. 103(a) as being unpatentable over Agrawal and further in view of Ho (US 6,810,377).

Regarding claims 12 and 23, Agrawal describes a method of (transmitted) digital media data stream comprising a stream of data elements (packets/cells), comprising:

[a jitter buffer to] (a) receiving the data stream (fig. 1 & col. 2, lines 50-52, stream receiver 30) and (b) hold each data element that is received prior to an end of a time period in a buffer until the end of the time period, at which time the data element is released for playout (col. 5, lines 24-33, buffer control 200 holds each data stream element in buffer slot designated by a respective pointer until timer 200, after T_r seconds, signals the element to be moved to output device 40);

[a controller to] (d) monitor a loss rate at which data elements in the data stream are not received by the end of their respective time periods (col. 6, lines 15-17, monitoring any changes to the PLR (packet loss rate) of the data stream in the buffer), and (e) adjust a duration of the time period based upon the loss rate (col. 6, lines 15-17, updating (adjusting) the buffer operating characteristics described in col. 5, lines 24-39, including adjusting the timer 230 to a new T_r seconds delay packet rate).

Agrawal fails to describe:

[a lost data element recovery mechanism to] (b) estimating, by an adaptive jitter buffer, a parameter of the unreceived data element based on received subsequent data element.

Ho describes:

receiving, by an adaptive jitter buffer, a subsequent data element that follows the unreceived data element in the data stream, and estimating, by an adaptive jitter buffer, a parameter of the unreceived data element based on received subsequent data element (col. 3, lines 30-33 & col. 4, lines 40-42, receiving a subsequent frame (data element) following a missing (unreceived) frame), and interpolating (estimating) the parameter of the missing frame).

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to describe estimating an unreceived data element based on the subsequent data element as per Ho for the teaching of Agrawal.

The motivation for combining the teaching is that it eliminates any unnatural sounding at the output (Ho, col. 3, lines 24-27).

Regarding claims 13 and 24, Agrawal and Ho combined describe that the receiving step (c) comprises receiving a plurality of subsequent data elements that follow the unreceived data frame (element) in the data stream and using Linear Predictive Coding (LPC) to estimate a parameter of the unreceived data frame based on a subsequent data element as per claim 12 (Ho, abstract), but fails to teach estimating the parameter of the unreceived data element based on the received subsequent data elements.

The examiner takes office notice that it is well-known in the art at the time of invention by applicant mathematically to linearly interpolate a missing data point (element) by using two subsequent data points in auto generating a chart/graph.

The motivation for using the two subsequent data points (elements) in generating a previous data point is so that a complete chart flow may be automatically generated even though initial datum/data may be missing.

Regarding claims 14 and 25, Agrawal fails to describe:

estimating a parameter of the unreceived data element based on the received subsequent data element and on a prior data element that precedes the unreceived data element in the data stream.

Ho describes:

estimating a parameter of the unreceived data element based on the received subsequent data element and on a prior data element that precedes the unreceived data element in the data stream (col. 3, lines 30-33 & col. 4, lines 40-42, receiving a previous and subsequent frame (data element) following a missing (unreceived) frame), and interpolating (estimating) the value (parameter) of the missing frame).

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to describe estimating an unreceived data element based on the subsequent data element as per Ho for the teaching of Agrawal.

The motivation for combining the teaching is that it eliminates any unnatural sounding at the output (Ho, col. 3, lines 24-27).

Regarding claims 21 and 30, Agrawal suggests that the time period begins for each transmitted data element when the data element is sent by a transmitting end (col. 5, lines 46-47, in using the timestamp of the sender).

Regarding claim 32, Agrawal and Ho combined describe all limitations set forth in claim 23. Agrawal further describes:

the media data stream is an encoded audio data stream comprising a plurality of audio data elements, each representing a porting of a transmitted audio session (col. 1, line 13, encoded audio data stream of audio data packets).

Regarding claim 33, Agrawal and Ho combined describe all limitations set forth in claim 23. Agrawal further describe that the data elements are frames of encoded data (col. 1, line 13, encoded audio/video data frames).

5. Claim 18-20, 22, 27-29 and 31 are rejected under 35 U.S.C. 103(a) as being unpatentable over Agrawal in view of Ho as applied to claim 16 above, and further in view of Yao.

Regarding claim 18, Agrawal and Ho combined describe all limitations in claim 17. Agrawal fails to describe:

adjusting step (e) comprises increasing the duration of the time period if the loss rate is above a first threshold.

Yao describes:

adjusting step (e) comprises increasing the duration of the time period if the loss rate is above LOSS_HYST (first threshold) (col. 5, lines 8-15, “On loss rate axis 310, a loss hysteresis threshold [LOSS_HYST] 312 defines a range 314 between LOSS_HYST and 1.0. In this range, an excess loss rate contributes to a decrease in transmission rate” [i.e. increased duration of time period].)

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to incorporate the monitoring and the adjustment of the playout using the loss rate as in Yao for the teaching of Agrawal and Ho.

The motivation for combining the teaching is that "The [loss rate] statistics provide indications of congestion of the data network" (Yao, col. 2, lines 56-57), and the statistic may be used to minimize such network congestion.

Regarding claim 19, Agrawal, Ho and Yao combined describe all limitations in claim 18. Agrawal further describes adjusting step (e) comprises:

increasing the duration of the time period by an amount that is substantially equivalent to a duration of the media represented by one data element if the loss rate is greater than the first threshold (Agrawal, col. 6, lines 25-29, repeating a data element 1x to increase time period equivalent to a data element).

Regarding claim 20, Agrawal, Ho and Yao combined describe all limitations in claim 18.

Agrawal fails to describe: decreasing the duration of the time period if the loss rate is below than a second threshold.

Yao describes adjusting step (e) comprises:

decreasing the duration of the time period if the loss rate is below than a second threshold (fig. 3, -LOSS_HYST #316) that is lower than the first threshold (LOSS_HYST) (fig. 3, where -#316 is lower than #312).

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to incorporate the monitoring and the adjustment of the playout using a second threshold as in Yao for the teaching of Agrawal.

The motivation for combining the teaching is that “The [loss rate] statistics provide indications of congestion of the data network” (Yao, col. 2, lines 56-57), and the statistic may be used to minimize such network congestion.

Regarding claim 22, Agrawal, Ho and Yao combined describe all limitations in claim 12. Agrawal further describe that the data elements are frames of encoded data (col. 1, line 13, encoded audio/video data frames).

Regarding claim 26, Agrawal and Ho combined describe all limitations set forth in claim 23. Agrawal fails to describe:

a controller monitoring a loss rate at which data elements in the data stream are not received by the end of their respective time periods.

Yao describes:

computing (monitoring) a loss rate at which data elements in the data stream are not received by the end of their respective time periods (fig. 2, period of sequence of packet dP=17, and col. 4, lines 29-31, “Rate controller 116 computes two statistics for such a sequence of sent packets 200. the first is a loss rate,”) and to adjust a duration of the time period (transmission rate) based upon the loss rate (col. 4, lines 59-61, “a rate controller computes an excess loss rate, L-L[0] and a loss ratio 1-L[s] in order to adjust the transmission rate”).

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to incorporate the monitoring and the adjustment of hold period before playout using the loss rate (unreceived data element) as per Yao for the combined teachings of Agrawal and Ho.

The motivation for combining the teachings is that “The [loss rate] statistics provide indications of congestion of the data network” (Yao, col. 2, lines 56-57), and the statistic may be used to minimize such network congestion.

Regarding claim 28, Agrawal, Ho and Yao combined describe all limitations in claim 27.

Agrawal fails to describe:

the controller is adapted to increase the duration of the time period by an amount that is substantially equivalent to a duration of the media represented by one data element if the loss rate is greater than the first threshold.

Yao describes:

the controller is adapted to increase the duration of the time period by an amount that is substantially equivalent to a duration of the media represented by one data element if the loss rate is greater than the first threshold (col. 4, lines 36-40, where the POB level is the buffer size and the increase of duration by a first amount is equivalent to the [extra] time in transmitting a byte of data [element]).

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to incorporate the monitoring and the adjustment of hold period

before playout using the loss rate (unreceived data element) as per Yao for the combined teachings of Agrawal and Ho.

The motivation for combining the teachings is that “The [loss rate] statistics provide indications of congestion of the data network” (Yao, col. 2, lines 56-57), and the statistic may be used to minimize such network congestion.

Regarding claim 29, Agrawal, Ho and Yao combined describe all limitations in claim 27. Agrawal fails to describe:

the controller is adapted to decrease the duration of the time period if the loss rate is below than a second threshold.

Yao describes:

the controller is adapted to decrease the duration of the time period if the loss rate is below than a second threshold (fig. 3, -LOSS_HYST #316) that is lower than the first threshold (LOSS_HYST) (fig. 3, where -#316 is lower than #312).

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to incorporate the monitoring and the adjustment of hold period before playout using the loss rate (unreceived data element) as per Yao for the combined teachings of Agrawal and Ho.

The motivation for combining the teachings is that “The [loss rate] statistics provide indications of congestion of the data network” (Yao, col. 2, lines 56-57), and the statistic may be used to minimize such network congestion.

Regarding claim 31, Agrawal and Ho combined describe all limitations in claim 23. Agrawal fails to describe:

a decoder adapted to receive data elements from the Jitter buffer and to decode the data elements to produce decoded data elements representing media samples.

Yao describes:

a decoder (network node #110B) adapted to receive data elements from the Jitter buffer (fig. 1, intermediate node #104) and to decode the data elements to produce decoded data elements representing media samples (fig. 1, where application layer #112 receiving decoded packets/elements from (lower) transport & network layers #118 and #120).

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to incorporate a decoder to decode data outputs from a jitter buffer (intermediate network node) as per Yao for the combined teachings of Agrawal and Ho.

The motivation for combining the teachings is that the data is transported from one network node to another via the standardized ISO protocol stack, requiring a layered approach to transmission of data (coding & decoding).

(10) Response to Argument

I. Rejections under Agrawal: Claim 1

From p. 6 last paragraph to p. 7 paragraph 1, the appellant argues that Agrawal does not teach the following claim language steps:

"(b) holding each data element that is received prior to an end of a time period associated with each data element in a buffer unit the end of the time period, at which time the data element is released for playout;

(c) monitoring a loss rate ..

(d) adjusting a duration of the time period based upon the loss rate."

In response to appellant's argument, the examiner respectfully disagrees with the argument above.

The examiner has already cited in the Office Action col. 5, lines 24-30 that timer 230 signals buffer control 200 the release of each buffered packet to output 250, corresponding to the claim language “holding each data element that is received “ and "the data element is released for playout”.

The examiner also understands that the timer 230 is a delay timer (see col. 3, line 18), where delay is understood in the art as a time period. Agrawal further explained in col. 6 lines 7-9, where buffer delay is defined as the time between receipt and output of each packet from Agrawal’s circuit system in fig. 1, the buffer delay is from the delay timer 230, which is controlled by control 10. The above mentioned (buffer) delay corresponds to the claim language of "a time period associated with each data element in a buffer".

Hence, Agrawal indeed describes a buffer holding each data element for a time equal to buffer delay before playout, thus fulfilling the claimed language step (b) as argued.

On p. 7 paragraph 2, the appellant also argues that Agrawal does not teach claim 1 step (d): “adjusting a duration of the time period based upon the loss rate."

In response to appellant's argument, the examiner respectfully disagrees with the argument above.

The examiner understands that Agrawal adjusts the buffer delay (duration of the time period): abstract & col. 4, lines 48-52: buffer delay is set (adjusted) according to the total end-to-end delay (TED), where, in addition, Agrawal describes “The TED is chosen to minimize the packet loss rate (PLR)”, col. 3, lines 34-37.

Hence, it is understood that the Agrawal's invention discloses that the buffer delay (duration of the time period) is set (adjusted) based upon the PLR (loss rate), thus fulfilling claim 1 step (d).

(11) Related proceedings appendix

No decision rendered by a court or the Board is identified by the examiner in the Related Appeals and Interferences section of this examiner’s answer.

Conclusion

For the above reasons, it is believed that the rejections should be sustained.

Respectfully submitted,

Warner Wong

November 20, 2000

Conferees:
/Chi H Pham/
Supervisory Patent Examiner, Art Unit 2416

12/18/08

/Kwang B. Yao/
Supervisory Patent Examiner, Art Unit 2416